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Kurt A. Summe 5/4/07

Kurt A. Summe, Reg. No. 36,023 Date

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P. O. Box 1450
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Sir:

ADDENDUM TO RESPONSE

TO OFFICE ACTION MAILED JANUARY 3, 2007

On May 3, 2007, Applicants filed a Response to the January 3, 2007 Office Action in the above-referenced case. In the Response, for the benefit of the Examiner, Applicants indicated an English translation of the Detailed Description of German Patent Number DT2628259, as

attached to the Response. In error, the translation was not attached, and is thus, attached to this document. The Response was a complete Response as filed and this translation is not necessary for the Response, but was being sent as a courtesy.

Applicants believe there is no charge for filing this Addendum. If any I fees are necessary, the Commissioner may consider this to be a request for such and charge any necessary fees to deposit account 23-3000.

Respectfully submitted,

WOOD, HERRON & EVANS, L.L.P.

Kurt A. Summe

Kurt A. Summe
Reg. No. 36,023

2700 Carew Tower
441 Vine Street
Cincinnati, Ohio 45202
(513) 241-2324
(513) 241-6234 facsimile
ksumme@whepatent.com

Fig. 2 Diagrams that show the connection between the individual – up to 6 - signals that occur in the circuit as shown in Fig. 1

Construction of a Talk-Back Circuit

The two participating stations 1a and 1b, each of which contains a microphone and a headset or loudspeaker L, are both connected to an input mixer with filter 2, to which an iterative network, a dynamic attenuator 3, a preamplifier 4 and a crossover network 5 are connected. The crossover network 5 splits the frequency spectrum offered to it into branch a and branch b. Branch b only contains the performance portion of the input signal out of one, for voice transmission deemed barely adequate frequency spectrum, of for example, 300 Hz to 3,000 Hz. Branch a contains all remaining frequencies whereby it is not deemed to be critical, however, if branch a still contains some frequencies in the speaking frequency range. The signal of branch a is fed to the iterative network from noise amplifier 6, AC / DC converter 7 and sample and hold circuit 8. The signal of branch b is fed to the iterative network out of speech filter 9, speech amplifier 10, a dynamic attenuator 11, a speech and mix filter 12, and a final amplifier 13. The final amplifier 13 is fed back to the speech and mix filter in frequency dependent mode, whereby the frequency response is dislocated with increasing strength to higher frequencies, in order to achieve an additional improvement of the understandability of speech. The outlet of speech amplifier 10 is connected with AC / DC converter 14 and a sample and hold storage circuit 15. The output of this

sample and hold circuit 15, together with the output of sample and hold circuit 8 is connected to a control signal mixer 16, that links the signals c and d out of the two sample and hold circuits 8 and 15 multiplicatively with a signal e that is fed to dynamic attenuator 3 as control signal. The output of final amplifier 13, whose output signal is labeled as f, is connected to both participant stations 1a, 1b and supplies the headsets or the loudspeakers L there.

Additionally, an AC / DC converter 17, whose output is connected with the input of a sample and hold circuit 18, is connected to final amplifier 13. The output of this sample and hold circuit 18, together with the output of sample and hold circuit 8, is connected with the control signal mixer 19 on the input side, whose output has a connection with dynamic attenuator 11. The two input signals g and h that are fed to control signal mixer 19 are multiplicatively linked to output signal i in control signal mixer 19. This signal i controls the degree of damping of the speech signal that runs through dynamic attenuator 11. The control signal g represents an inverted value of the speech level stored in sample and hold circuit 18, while control signal h represents a non-inverted value of the noise level stored in sample and hold circuit 8. Through this it is achieved that the output capacity of final amplifier 13 is raised with increasing noise level, while corresponding damping of the transfer channel counteracts a loud mode of speech.

In addition to dynamic attenuator 11, a threshold value circuit 20 whose outlet is connected with the inlet of control unit 21, is connected to amplifier 10. The output

of this control unit 21 is connected with an inlet of sample and hold circuit 8 and an inlet of dynamic attenuator 11.

To the circuits described up to now, a further circuit unit is connected that comprises a priority encoder 22, an interfering voltage mixer 23, a dynamic attenuator 24 and a control unit 25. The dynamic attenuator 24 has three inlets that are connected with the outlets of control unit 21, with interfering voltage mixer 23 and sample and hold circuit 8. The two outlets of priority encoder 22 are connected to the inlets of control unit 21 and interfering voltage mixer 23. The three inlets of control unit 25 are connected with the outlet of control unit 21, the output of sample and hold circuit 8 and the output of priority encoder 22. The priority encoder 22 is fed recognition signals C from radio transmitting or radio receiving sets on the input side, while the input of interference signal mixer 23 receives the wanted signals E directly from radio receiving or radio transmitting units. Additional devices for various purposes can be connected to control unit 25 with output signal D.

In the application of motorcycle driver and passenger that was addressed several times already, the two participating stations 1a, 1b, that consist of a microphone unit M and a headset L, are located respectively in the crash-helmet. The remaining portion of the shown circuit is located in a small box that is connected with the participant stations via cable with desired break points, and can, for example, be worn in the protective waist belt of the driver and is connected with the battery of the motor cycle. In this small box suitable plug-in connections such as radio

transmitting, radio receiving and similar [devices] can be connected. For other applications for which a cable connection between the individual participant stations is not possible, for example in the application 'parachutist' the connections that can not be realized with cables are to be replaced with radio connections, whereby it is very apparent for an expert that the presented circuit schematic is to be complemented with corresponding transmitting and receiving components and power supply.

Functional Description

The output signal of the two microphone units M are added up in opposite phase in input mixer and filter 2 so that essentially only the differential of the wanted signal is passed on to dynamic attenuator 3. The filter dedicated to input mixer 2 serves to block high frequency interfering signals at the inlet. The wanted signal that leaves input mixer 2 in which speech and noise still have equal rights, is fed to crossover network 5 via dynamic attenuator 3 and preamplifier 4. The signal chain formed by input mixer 2, attenuator 3 and preamplifier 4 has a wide bandwidth so that all noise frequencies that are to be considered are captured. Crossover network 5 divides the signal that is offered to it into an output signal a and an output signal b, whereby the output signal b represents that component of the input signal that has a frequency band from 300 Hz to 3,000 Hz. Output signal a contains the remaining frequency band of the input signal. Through noise amplifier 6, the noise signal a is raised to a level that takes advantage of all of the dynamic area of noise amplifier 6 and the AC / DC converter 7 that is connected to it. The direct current supplied

by AC / DC converter 7 is continually sampled in the speech pauses by sample and hold circuit 8 that is connected with it. If speech takes place, the value that was determined at the end of the last pause in speech remains stored during the entire speech segment that follows. The switching of sample and hold circuit 8 from continuous sampling operation during pauses in speech to permanent storage operation in which the value determined at the end of the pause in speech is 'frozen', is brought about through signal 1 from control unit 21 that is controlled by threshold value circuit 20, in which the end of pauses in speech is determined, which will be explained in more detail below. Only pure noise levels that are not contaminated with speech components are stored in sample and hold circuit 8 on account of this operating mode. Because in the usual operation of talk-back systems speech and pauses in speech alternate in rapid succession, noise levels that change with time in the individual pauses in speech are also continually captured and considered.

Out of the available frequency spectrum, the frequency band between 300 Hz and 3,000 Hz that is required for speech transmission is filtered out in crossover network 5 for signal branch b. In subsequent speech filter 9, the frequencies that are required to make speech understandable are additionally singled out via narrow-band filters. The filter losses are compensated again with speech amplifier 10. Via dynamic attenuator 11 and strengthening-dependent filter 12, the speech signal reaches final amplifier 13. This performance amplifier 13 supplies the loudspeakers or the headsets L of the two participant stations 1a and 1b in parallel manner.

The speech signal that is enhanced through speech amplifier 10 is fed to threshold value circuit 20 whose response threshold is controlled for enlargement of the interfering voltage intervals

between noise and speech through output signal h of the sample and hold circuit. The higher the noise level, the higher the trigger threshold of threshold value circuit 20 is set. Threshold value circuit 20 serves as speech recognition switch that takes on the first switching state at times when speech takes place and takes on the second switching state in speech pauses. For this reason, threshold value circuit 20 must separate speech from noise components that are in the same frequency band as speech and be able to distinguish between them. As in many cases of applications the frequency components that occur in the environmental noise are in the speech frequency band and could, at their volume, easily exceed the volume of natural speech, the solution of this problem is very difficult. In accordance with experience, one speaks even louder in practice depending on the level of environmental noise. It became evident for all noise levels that occur in practice that the speech performance offered by the speaker in the speech frequency band of 300 Hz to 3,000 Hz is always larger than the performance portion of the noise in the same frequency band. This insight is being used for clean speech recognition in the case at hand in that one raises the threshold value in threshold value circuit 20 dependent on signal h of sample and hold circuit 8 with rising noise levels. Threshold value circuit 20 contains a re-triggerable mono-stable trigger circuit whose running time is aligned with the time constant of the lowest speech frequency, here 300 Hz. The re-triggerable mono-stable trigger circuit is a mono-stable trigger circuit that remains switched after each occurring switching impulse flank for a predetermined amount of time, even if this switching impulse flank occurs at a time at which the mono-stable trigger circuit is already switched. Expressed in other words, the mono-stable trigger circuit remains switched for a predetermined time after the last occurring switching

impulse flank, regardless whether it was switched at the point in time of the occurrence of the switching impulse flank or not. Short pauses that are characteristic in human speech and their individual sounds are consequently not interpreted as pauses in speech.

In the circuit shown in Fig. 1, separate capture of speech and noise is used to switch dynamic attenuator 11 through signal 1 via threshold value circuit 20 and control unit 21 that is connected with it to maximum damping, whereby no noise is transmitted to headset L in the pauses of speech that were determined. A damping of approximately 70 dB is sufficient here; it is practically equal to the interruption of the transmission channel. A disturbance through increased noise components in the speech pauses that occur in generally known talk-back systems is thus avoided. While the switching off and switching on of dynamic attenuator 3 at the beginning and end of a speech segment can be solved technically so that none or only very soft snapping sounds occur, this is however, not very simple because of the very quick on-switching at the beginning of the speech segment. The following solution is technically simpler and consists of an interruption of the transmission channel through dynamic attenuator 3 only when a pause in speech takes place that is more than, for example, 3 seconds. Pauses of less than 3 seconds as they occur, for example, between two spoken sentences or words, are then not interpreted as pauses in speech, whereby the switching frequency is significantly reduced. The switching noises that then still occur are then no longer perceived as irritating. In order to achieve this mode of functioning, control unit 21 is provided that emits signal k which introduces a pause in speech from threshold value circuit 20 at the outlet delayed by about 3 seconds, while signal k that identifies the end of a pause in speech appears without delay at the outlet of control unit 21.

Control signal mixer 16 and 19 serve to set dynamic attenuator 3 and 11 at the beginning of the speech segment already to an amplification factor that is adapted to the speech level that is to be expected. Since speech levels before and after short pauses in speech most often do not differ significantly from one another, this 'pre-programming' is successful in a simple manner with the shown circuit in that one determines the speech level directly before a pause in speech, retains it during the pause in speech and uses this speech level for pre-programming of the amplification factor of dynamic attenuator 3 or 11 at the beginning of the subsequent speech segment. For this purpose, the speech level determined at the end of a speech segment is retained in sample and hold circuits 15 and 18, that receive direct current from the AC / DC converters 14 and 17 located before them whose amplitudes are determined by the respective speech level, that store the last speech level that was determined before a pause in speech during the following pause in speech. It is essential here that sample and hold circuits 15 and 18 are switched immediately after the end of each speech syllable to permanent storage operation and not only at the actual end of the speech segment that is usually lengthened by control unit 21 by 3 seconds. For this reason, sample and hold circuits 15 and 18 are controlled by signal k from threshold value circuit 20 directly, and not by signal 1 of control unit 21. If one were to use signal 1 for switching of sample and hold circuits 15 and 18, the pause that occurs at the end of a speech segment of 3 seconds would be interpreted as quiet speech and consequently, a speech level that is too low for the pre-programming of dynamic attenuators 3 and 11 would be used.

The circuit shown in Fig. 1 contains three control circuits I, II, and III, that are functionally aligned with each other and that influence each other at least in part. Control circuit I is formed by dynamic attenuator 3, preamplifier 4, crossover network 5, noise amplifier 6, AC / DC converter 7 and sample and hold circuit 8. Control circuit II contains a dynamic attenuator 3, preamplifier 4, crossover network 5, speech filter 9, speech amplifier 10, AC / DC converter 14 and sample and hold circuit 15. Control circuit III finally contains AC / DC converter 17, sample and hold circuit 18, control signal mixer 19 and dynamic attenuator 11. The function of these three control circuits are explained in more detail in the following.

Control signal mixer 16 through its output signal e, shall control dynamic attenuator 3 in such a way that at the outlet of preamplifier 4, a signal as constant as possible is created, in order to protect the circuit that is connected to it from over-modulation. For this purpose, control signal mixer 16 is fed as input the output signals c and d of sample and hold circuits 8 and 15 that are multiplicatively linked with one another in the control signal mixer in order to form the output signal e. As a consequence, control signal mixer 16 undergoes an adaptation of the amplification factor of dynamic attenuator 3 in the speech segments to the changing speech level for frozen noise levels, while the speech level during pauses in speech is frozen in sample and hold circuit 15 and the amplification factor is constantly being adapted to the changing noise level. On account of the multiplicative linking of the two signals c and d to control signal e it is achieved that the degree of damping of dynamic attenuator 3 rises for rising noise levels as well as for rising speech levels. To this, the following example:

Under the pre-condition that no noise is present and that speech is naturally soft, sample and hold circuit 15 with its output signal d reports amplitude that is too small for speech amplifier 10. The control deviations thus formed increases the amplitude of dynamic attenuator 3. With increasing speech volume, the input signal d of sample and hold circuit 15 also rises. The consequence is a reduction of the amplitude factor through dynamic attenuator 3. In order to secure the entire system from over-modulation, it is sufficient to provide a dynamic area of 65 dB in dynamic attenuator 3. In the further example it is to be assumed that a maximum noise level is present. The noise components pass dynamic attenuator 3 as well as the speech components. Naturally, with an increasing noise level, speech also becomes louder. The amplification factor of dynamic attenuator 3 is essentially determined by the output current of speech amplifier 10 and thus also the amplification of the noise component. Should the noise component become so strong during the pauses in speech that the noise measuring channel that consists of noise amplifier 6, AC / DC converter 7 and sample and hold cireuit 8 would be over-modulated and thus could no longer behave as pre-programmed, signal c via control signal mixer 16 would function in a damping mode in like manner to output signal d of sample and hold circuit 15. The consequence of this is that the amplification factor is continually pre-programmed through the noise level for speech.

The circuit at hand also takes into account the following special situation. A drive on a motorbike is to be assumed that accelerates from a standing position. While standing, only the motor noises are present that are strongly damped through the crash-helmet so that the

noise level that is determined is correspondingly low. If speech takes place during acceleration, the noise measuring channel remains blocked for the entire acceleration phase. If speech is not initiated until a speed is attained at which the noise level is already as large as the speech level without noise, the system remains interconnected in the subsequent pause in speech because threshold value circuit 20 considers the noise components that lie in the speech frequency band as speech. However, it is possible without activating switches to switch the entire system back to 'normal operation' by simply yelling into microphone M, and thus feeding speech branch b a very high speech level. Through this high speech level, the amplification factor of dynamic attenuator 3 is reduced so much via AC / DC converter 14, sample and hold circuit 15 and control signal mixer 16 of, that the noises are damped automatically. With that, the noise components in threshold value circuit 20 are again under trigger level and thus release the noise measuring channel for measuring again. Dynamic attenuator 11 is switched to the maximum degree of damping and no signal is transmitted in the following pauses in speech.

Control circuit III serves to raise the loudness of output signals of final amplifier 13 with increasing noise levels. AC / DC converter 17 with sample and hold circuit 18 ensure that performance-appropriate maladjustments to final amplifier 13 are ruled out and over-modulation is avoided. This demand requires that control signal g must be offered to control signal mixer 19 inversely to control signal h. The linking in control signal mixer 19 also takes place in a multiplicative fashion. The output performance of final amplifier 13 is raised with increasing noise level until the limit of the duty cycle of final amplifier 13 has been reached.

With the aid of the diagrams presented in Fig. 2 to 6, the control characteristics of the three control circuits I, II, and III and their reciprocal influences are shown in detail.

The speech level identified by both participant stations 1a and 1b produce via functional blocks 2, 3, 4, 5, 9, 10, 14, and 15 the output signal d. It can be seen in Fig. 2 that with a rising speech level, signal d at first remains constant up to point n and then decreases. Through this, the stabilizing effect related to the output amplitude of speech amplifier 10 is achieved. At point x in the speech level diagram of Fig. 2, the amplification factor equals 1, i.e. the voltage given off by microphone M corresponds with the output voltage of speech amplifier 10. With an increasing speech level, a damping effect acts on dynamic attenuator 3 that holds the output amplitude of speech amplifier 10 constant. Under the value x, dynamic attenuator 3 acts strengthening and thus increases the weak input signals to the pre-programmed constant level.

Fig. 3 shows the dependence of signals c on noise level a. Up to input noise level y, signal c corresponds to a multiplicand corresponding to factor 1. Expressed in other words this means that up to this point y, the signal d is fully responsible for the amplification factor. The output signal e from control signal mixer 16 follows the equation e = c x d. Since factor c equals 1 up to point y, e automatically becomes d. The danger of over-modulation of the noise measurement channel exists only for increasing noise levels starting with noise level y. For this reason, signal c acts as damping factor upon dynamic attenuator 3 starting with point y. As a

consequence, dynamic attenuator 3 already has a pre-programmed damping factor when the speech level arrives, before the speech level is captured by sample and hold circuit 15.

The combination out of the two multiplicatively linked signals c and d is shown in Fig. 4. The spatial arrangement shows the influence of the speech level as dependent upon the noise level. Up to point o, the output voltage of speech amplifier 10 rises approximately linear with the speech level of 1a or 1b. Without noise level the value U2 corresponds to the value U1 and thus characterizes the stabilizing effect of the first control circuit I. In spite of increasing noise level the shape of the curve does not change up to point p. Point p in Fig. 4 corresponds to the assigned noise level value y in Fig. 3. With increasing noise level, the voltage at U1 reduces to the value at U1S. With increasing speech level, however, the value rises to U2 at maximum input voltage at the inlet of input mixer 2. Further, above the noise level, the progression of the trigger threshold of threshold value circuit 20 is drawn in. At a noise level of zero the beginning threshold corresponds to the value q. The trigger threshold also rises with increasing noise level. Beginning with point p, the stabilizing effect of control circuit II appears that maintains the noise level at a nearly constant level. That is why the progression of the curve of the trigger threshold becomes flatter in correspondence with the noise level (see point r).

In the additional coordinate system that is drawn on the noise level axis, the progression of output signal h of sample and hold circuit 8 as dependent on the noise level becomes clear. With a rising noise level the amplitude of this signal h rises up to point s. Point s corresponds with the

use of a control circuit in accordance with its previously set desired voltage US. With increasing noise level, a stabilizing effect that also relates to signal h, appears.

In Fig. 5, the dependence of output performance of final amplifier 13 on noise level is shown. Even at a noise level of zero, an output capacity must be present as a matter of course. This value is determined in such a way that with the adjustment possibilities in the helmet the basic volume level can be set according to the respective ability to hear. With increasing noise level, the output capacity rises up to point Z, which corresponds to noise level y in Fig. 3. Beginning with that point, a stabilizing effect appears on account of control circuit II (noise channel). The maximum attainable theoretical output capacity through the noise level would exceed the performance available to final amplifier 13. This condition is avoided through control circuit III that is comprised out of functional units 17, 18, and 19.

Fig. 6 shows the influence of control circuit III. With increasing amplitude of signal h, the output performance increases. The desired value that is pre-programmed as value W in control signal mixer 19 determines the control operation of control circuit III. Starting with this point, the output performance is limited to such a value that over-modulation is ruled out. The linear part of point O up to point W is set by parameter h as desired value. The task of the control circuit consists of keeping the output performance at the required value h independent of changes in the voltage supply or other interfering influences.

Because of this, control circuits I, II and III can be separated into two functional units that are independent of one another. Control circuits I and II ensure that the entire system is secured against over-modulation and that constant outlet amplitude is present at the outlet of speech amplifier 10. Control circuit III has the task to control the output performance as dependent on the noise level and to maintain it independent of interfering influences. The voltage peaks that occur for a short time after switching on would inevitably drive the final amplifier to saturation. This over-modulation is also avoided through control circuit III.

Following, the function of circuit pieces 22 to 25 as shown in Fig 1 are explained in more detail. If no speech takes place, control unit 21 emits signal m to dynamic attenuator 24 indicating release for interconnection of interfering voltage E to mixer stage 12. Dynamic attenuator 24 is subsequent to interference signal mixer 23 that mixes external sources of tone dependent on the decision of priority encoder 22. The loudness of these signals is also controlled by noise level h, which is fed to dynamic attenuator 24.

A summary of the control and control signals for eventual external additional devices takes place in control unit 25 that functions simultaneously as input and output interface.

The following application example shall explain the function of priority encoder 22 and the switching steps assigned to it. Both participant stations 1a and 1b were placed in the crash-helmets of a motor cyclist and his passenger. A radio transmitter and receiver are connected to

priority encoder 22 and interfering voltage mixer 23. As long as no speech takes place the radio conversation is blended in. With the first syllable that is spoken, this signal is blended out and the speech is interconnected with final amplifier 13. If a traffic announcement, usually introduced by an identifying signal, takes place during speech, it is recognized as such by priority encoder 22, and priority encoder 22 gives notice of such through a special signal tone that is mixed with the speech transmitted. If no speech takes place, this signal tone appears by itself as a matter of course. It is up to the speech partners to accept this information. If the speech connection is terminated, the traffic announcement goes to headsets L. If the conversation continues, this announcement is suppressed. A received radio message, however, interrupts the internal speech connection and switches the receiving signal to final amplifier 13 for the duration of the radio message.

- Patent Claims -